

WHAT IS CLAIMED IS:

1. A method for approximating  $y(n) = 1/x(n)$  in FM demodulation, where  $x(n) = I^2(n) + Q^2(n)$ , comprising:

- (a) receiving a prior estimated value of  $1/x(n)$ ;
- (b) receiving a present value of  $x(n)$ ;
- (c) adjusting the prior estimated value of  $1/x(n)$  to compensate for an error between the prior estimated value of  $1/x(n)$  and the present value of  $1/x(n)$ ; and
- (d) outputting the adjusted prior estimated value of  $1/x(n)$  as the present value of  $1/x(n)$ .

2. The method of claim 1, wherein the prior estimated value of  $1/x(n-1)$  equals  $1/(I^2(n-1) + Q^2(n-1))$ , wherein  $I(n)$  is an input signal and  $Q(n)$  is a quadrature-phase signal of the input signal  $I(n)$ .

3. The method of claim 2, wherein the input signal  $I(n)$  comprises a band pass filtered secondary audio program signal.

4. The method of claim 1, wherein the present value  $x(n)$  equals  $I^2(n) + Q^2(n)$ , and wherein  $I(n)$  is an input signal and  $Q(n)$  is quadrature-phase signal of  $I(n)$ .

5. The method of claim 4, wherein the input signal  $I(n)$  comprises a band pass filtered secondary audio program signal.

6. The method of claim 1, wherein an error signal equals  $(1 - x(n)y(n-1))a$ , wherein  $x(n) = I^2(n) + Q^2(n)$ ,  $y(n-1) = 1/(I^2(n-1) + Q^2(n-1))$ ,  $I(n)$  is an input signal,  $Q(n)$  is a quadrature-phase signal of the input signal  $I(n)$ , and "a" is a scaling coefficient!

7. The method of claim 6, wherein the input signal I(n) comprises a band pass filtered secondary audio program signal.

8. The method of claim 1, wherein the Y(n) signal equals  $y(n-1) + (1-x(n)(y(n-1))a$ , wherein  $x(n) = I^2(n)+Q^2(n)$ ,  $y(n-1) = 1/(I^2(n-1)+Q^2(n-1))$ , I(n) is an input signal, Q(n) is a quadrature-phase signal of the input signal I(n), and "a" is a scaling coefficient.

9. The method of claim 8, wherein the input signal I(n) comprises a band pass filtered secondary audio signal.

10. A method for demodulating an FM signal FM(n) from a secondary audio program signal, comprising:

(a) receiving in-phase I(n) and quadrature-phase Q(n) portions of the FM(n) signal

(b) generating a first portion of the FM(n) signal that is equal to  $I(n)Q(n)-I(n)Q(n)$ ;

(c) determining a value z(n) based on the first portion of the FM(n) signal;

(d) generating a second portion of the FM(n) signal that is equal to  $1/I^2(n)+Q^2(n)$ , wherein  $I^2(n)+Q^2(n)$  is equal to x(n) and  $y(n)=1/x(n)$ ;

(e) generating a value for y(n) based on  $1/x(n)$  that equals  $y(n-1) + (1-x(n)y(n-1))a$ ; and

(f) multiplying the z(n) value and the y(n) value to produce the FM(n) signal.

11. A system for approximating  $y(n) = 1/x(n)$  in FM demodulation, where  $x(n) = I^2(n) + Q^2(n)$ , comprising:

means for receiving a prior estimated value of  $1/x(n)$ ;

means for receiving a present value of  $x(n)$ ;

means for adjusting the prior estimated value of  $1/x(n)$  to compensate for an error between the prior estimated value of  $1/x(n)$  and the present value of  $1/x(n)$ ; and

means for outputting the adjusted prior estimated value of  $1/x(n)$  as the present value of  $1/x(n)$ .

12. The system of claim 11, wherein the prior estimated value of  $1/x(n-1)$  equals  $1/(I^2(n-1) + Q^2(n-1))$ , wherein  $I(n)$  is an input signal and  $Q(n)$  is a quadrature-phase signal of the input signal  $I(n)$ .

13. The system of claim 12, wherein the input signal  $I(n)$  comprises a band pass filtered secondary audio program signal.

14. The system of claim 11, wherein the present value  $x(n)$  equals  $I^2(n) + Q^2(n)$ , and wherein  $I(n)$  is an input signal and  $Q(n)$  is quadrature-phase signal of  $I(n)$ .

15. The system of claim 14, wherein the input signal  $I(n)$  comprises a band pass filtered secondary audio program signal.

16. The system of claim 11, wherein an error signal equals  $(1 - x(n)y(n-1))a$ , wherein  $x(n) = I^2(n) + Q^2(n)$ ,  $y(n-1) = 1/(I^2(n-1) + Q^2(n-1))$ ,  $I(n)$  is an input signal,  $Q(n)$  is a quadrature-phase signal of the input signal  $I(n)$ , and “ $a$ ” is a scaling coefficient.

17. The system of claim 16, wherein the input signal  $I(n)$  comprises a band pass filtered secondary audio program signal.

18. The system of claim 11, wherein the  $Y(n)$  signal equals  $y(n-1) + (1-x(n))(y(n-1))a$ , wherein  $x(n) = I^2(n)+Q^2(n)$ ,  $y(n-1) = 1/(I^2(n-1)+Q^2(n-1))$ ,  $I(n)$  is an input signal,  $Q(n)$  is a quadrature-phase signal of the input signal  $I(n)$ , and "a" is a scaling coefficient.

19. The system of claim 18, wherein the input signal  $I(n)$  comprises a band pass filtered secondary audio signal.

20. A method for approximating  $y(n)=1/x(n)$  in FM demodulation, where  $x(n)=I^2(n)+Q^2(n)$ , comprising:

- (a) receiving  $1/x(n-1)$ ;
- (b) receiving  $x(n)$ ;
- (c) adjusting  $1/x(n-1)$  to compensate for an error between  $1/x(n-1)$  and  $1/x(n)$ ; and
- (d) outputting the adjusted  $1/x(n-1)$  as  $1/x(n)$ .